Partial Reliable TCP

Yao-Nan Lien and Ming-Han Wu
Computer Science Department ,National Chengchi University , Taipei, Taiwan, R.O.C.
lien@cs.nccu.edu.tw

ABSTRACT-Some new information services over IP-based networks such as video streaming and VoIP (Voice over IP) can tolerate some packets lost in transmission without too much damage to their quality. The content carried in the packets of these services is not equally important in their replay processes. Unfortunately, the two most popular transport protocols, UDP and TCP, treat all packets equally without any discrimination. This paper proposes a new TCP protocol, named Partial-Reliable TCP (PR-TCP), which applies selective retransmission strategy to provide delivery guarantee to the selected packets designated by the application programs. In this way, we can save bandwidth consumption and reduce average delivery time without significant quality degradation. Two versions are proposed: Basic PR-TCP and Single-Side PR-TCP. Basic PR-TCP requires both ends of a connection to adopt PR-TCP while Single-Side PR-TCP only requires the sender end to adopt it.

We use NS-2 network simulator to evaluate PR-TCP against TCP Reno, TFRC and UDP. Two video stream samples are used for video sources. The simulation shows that PR-TCP can have better performance under various conditions.

***Keywords**:* TCP, MPEG

I. INTRODUCTION

As computer and communication technology advanced, many new information services over IP-based networks such as video streaming and VoIP (Voice over IP) are growing rapidly. These services can tolerate some packet lost without too much damage to their quality. The content carried in the packets of these services is not equally important in their replay processes. For example, key frames (e.g. I-Frames) of a video encoded in MPEG format are more important than others [10,11]. The loss of I-frames may have a large impact to the quality of the transmitted video, while the loss of other types of frames may only have a nominal damage. Unfortunately, the two most popular transport protocols, UDP [6] and TCP [5], treat all packets equally without any discrimination. TCP guarantees the delivery of all packets, while UDP doesn't. TCP may waste too much resource to guarantee the delivery of unimportant packets, while UDP may fail to deliver many important packets. Furthermore, UDP doesn’t have any mechanism to prevent itself from congesting the network when network bandwidth is shrunk.

We propose a new TCP protocol, named Partial-Reliable TCP (PR-TCP), which provide delivery guarantee to the selected packets designated by the application programs. In this way, we can save bandwidth consumption and reduce average delivery time without significant quality degradation. Furthermore, the congestion control mechanism is preserved in PR-TCP such that it can avoid speeding in transmitting packets. In fact, if the delivery of an object requires a stringent delivery time, the reduction of average delivery time may also lead to the reduction of overdue packets. Two versions are proposed: Basic PR-TCP and Single-Side PR-TCP. Basic PR-TCP requires both ends of a connection to adopt PR-TCP while Single-Side PR-TCP only requires the sender end to adopt it. It is much easier to deploy Single-Side PR-TCP on the client-server systems where only servers need to use PR-TCP.

The rest of this paper is organized as follow. In Section II, we review the relative background regarding to TCP and MPEG. We introduce our Partial Reliable TCP in Section III and evaluate it against others by simulation in Section IV. Finally, we conclude our main contribution of this paper and highlight some future work in Section V.

II. RELATED WORK

2.1 TCP

TCP (Transmission Control Protocol) is a connection-oriented reliable protocol [5]. It provides a reliable transport service between pairs of hosts (end nodes or terminal nodes) using the network layer service provided by the IP protocol. TCP enables two hosts to establish a connection and exchange streams of data. It guarantees the delivery of all data packets and also guarantees that packets will be delivered in the same order in which they were sent.

Packets may be lost in the transmission path due to various reasons such as network congestion and radio channel error. TCP adopts a complicate retransmission protocol to guarantee the delivery of packets. TCP modules reside at both ends of a connection which may have quite a few intermediate nodes in between. Thus, the source node of a TCP connection must determine the data rate based on its own poor knowledge of network status.

Because most hosts on the network do not have a good knowledge of the network status, it is impossible for them to have a perfect control on the speed they inject data into the network. Therefore, network will be congested from time to time. Network elements including routers and end terminals need to work hard to avoid network congestion. The congestion control within a TCP plays a critical role in adjusting data rate to avoid network congestion. Based on some window-adjustment algorithm, a TCP not only guarantees the successful packet delivery, but also maintains an "appropriate" data rate [3].

2.2 UDP

User Datagram Protocol (UDP) is another transport layer protocol [6], which does not guarantee reliability or ordering of datagrams (packets) in the way that TCP does. Packets may arrive out of order, appear duplicated, or go missing without notice. Avoiding the overhead of packet delivery guarantee makes UDP faster and more efficient. Time-sensitive applications often use UDP because dropped packets are usually preferable to delayed packets. However, UDP doesn't have any mechanism to adjust data rate to accommodate the change of network bandwidth. Once the network bandwidth gets shrunk, it continues to inject packets into the network. Overflowed packets may not only get themselves lost, but also waste network resources. Furthermore, UDP has no capability to provide selective packet protection for those applications that need it. To some applications such as real-time video streaming, dropping some packets may have a significant impact to the video quality. UDP may not be adequate for them.

2.3 Video Streaming Service

Video streaming, the technology used in many on-line video services such as YouTube, is becoming a critical Internet technology nowadays. A video streaming service is probably required to deliver concurrent video streams to a large number of users. To best use network bandwidth, most video streams are compressed. Data segments, (*frames*), may be correlated such that the loss of a frame may hurt the decodability of another frame. Therefore, frames are not equally important in term of impact to the quality of replayed video. MPEG is one of the most popular video encoding standards [10,11]. We use MPEG encoded video streams as an example to illustrate the usability of our PR-TCP.

MPEG video compression is based on motion compensated predictive coding with an I-B-P-frame structure as depicted in Fig. 1. An *I-frame* (Intra-coded pictures), also called a *key frame*, can be decoded independently without referring to other frames. To decode a *P-frame* (Predictive coded pictures), the previously encoded I-/P-frames need to be decoded first. To decode a *B-frame* (Bidirectionally predictive pictures), the frames that are before and after this B-frame need to be decoded first.

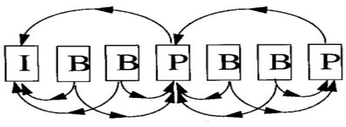


Fig. 1. I-P-B Frame Structure of MPEG

2.4 Summary

In summary, some real-time multimedia network services have stringent delay constraint, can tolerate some packet lost, and need a non-uniform data protection plan. A good transport protocol for these services must be able to provide selective data protection as well as to control data rate to accommodate the change of network bandwidth. Neither TCP nor UDP can fulfill this demand. TCP guarantees the delivery of data packets in a non-discrimination basis. It can also adjust data rate to accommodate the change of network bandwidth. For the services mentioned above, 100% guarantee for packet delivery is not only unnecessary, but also a waste of network resource.

On the other hand, UDP doesn't provide any data protection at all. It doesn't perform data rate adjustment either. Therefore, there is a great need for a new transport protocol for real-time multimedia network services. Although STCP provides selective protection mechanism, it is not as popular as TCP yet.

III. PARTIAL RELIABLE TCP

PR-TCP is designed to have all the capability of TCP and the selective packet protection functionality. When the upper layer software module hands a data segment to PR-TCP, it also designates a protection class for the data. PR-TCP then delivers the data segment to the destination in the form of packets according to the designated protection class. Since congestion control mechanism is highly dependent on the success or failure of packet delivery, the complicate congestion control mechanism need to be modified accordingly. We propose two versions of PR-TCP: Basic PR-TCP and Single-Side PR-TCP. Basic PR-TCP requires both ends of a connection to adopt PR-TCP while Single-Side PR-TCP only requires the sender end to adopt it. It is more difficult to deploy Basic PR-TCP, since both ends of a TCP connection has to execute the same version of TCP. On the other hand, Single-Side TCP is much easier to deploy since the receiver end can use any other version of TCP. For instance, in a client-server environment, only server needs to execute PR-TCP.

3.1 Basic PR-TCP

Basic PR-TCP supports three different classes of protection: (a) **Regular:** no protection; (b) **Certified:** protected up to a time limit; (c) **Registered:** protected without any time limit. It is not difficult to figure out the retransmission mechanism to support all three protection classes. Certified class is useful for those applications that have a stringent delay time constraint and to discard overdue packets.

The header format is modified to include a three-attribute *protection class* field: (a) **pt:** protection class of this packet; (b)**Bpt:** protection class of previous packet; (c)**Npt:** protection class of next packet.

To support Certified protection class, a parameter, *Retransmission Limit* (RL), is set to indicate the maximum number of permitted retransmissions. The value of RL can be calculated by dividing the life time of a packet, which is given by the calling application, by RTT (Round Trip Time). Once the number of retransmissions reaches the RL of a packet, the sender terminates the retransmission of a Certified packet and sends a FNP (Forward Next Packet) message to notify the receiver. Upon receiving a FNP message, the receiver stops waiting for the packet.

We reuse the congestion control mechanism of TCP NewReno [1] with a slight modification: Fast Retransmission is replaced with a Fast Selective Retransmission. State transition is shown in Table 1.

Table 1. State Transition of Basic PR-TCP
(SS: Slow Start, CA: Congestion Avoidance, FF: Fast Selective Retransmit and Fast Recovery)

Event	Current State	Next State	Action
new ACK	(SS)	(SS)	CWND=CWND*2
Timeout	(SS)	(SS)	threshold=(1/2)*CWND, CWND=1
CWND>= threshold	(SS)	(CA)	CWND=CWND+1
new ACK	(CA)	(CA)	CWND=CWND+1
Timeout	(CA)	(SS)	CWND=1
packet lost/Timeout	(CA)	(FF)	threshold = CWND*(1/2), CWND = threshold
non-Regular packet lost	(FF)	(FF)	No change
new ACK/Regular packet lost/, Certified RL==0	(FF)	(CA)	CWND=CWND+1
Timeout	(FF)	(SS)	CWND=1

3.2 Single-Side PR-TCP

For easy deployment, we also designed Single-Side PR-TCP, in which, only the sender side needs to adopt PR-TCP and the receiver side can use any other version of TCP. Single-Side PR-TCP supports two protection classes: (a) **Regular:** no protection; (b) **Registered:** protected without any time limit.
Single-Side PR-TCP doesn't support Certified protection class because it needs the cooperation of receiver side TCP.

The biggest challenge is to handle the loss of Regular packets. The receiver side TCP will be hung up waiting for the lost Regular packets forever. To solve this problem, the sender side sends out header replicas of Regular packets, called *pseudo packets*, automatically to fool the receiver side TCP. The probability of packet loss will be greatly reduced as shown in Fig. 2. Triple replication will be the best under most cases. Replication overhead is nominal since the size of a TCP header is only 20 bytes.

If a pseudo packet arrive the receiver before its real packet, the pseudo packet will be kept by the receiver and the real packet will be discarded. To prevent this from happening frequently, pseudo packets are delayed by at least one packet as shown in Fig. 3. In Fig. 3, packet 1 and 3 are Regular class. Replicas of packet 1 are sent after packet 2 and 3.

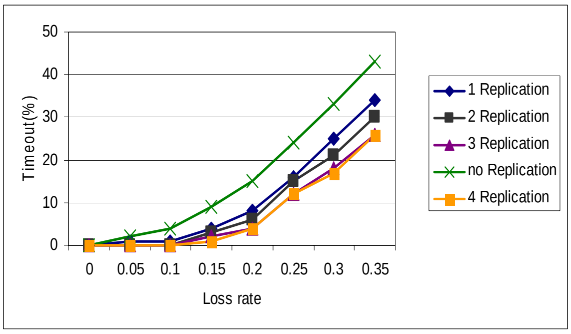


Fig. 2. Replications vs. Loss Probability

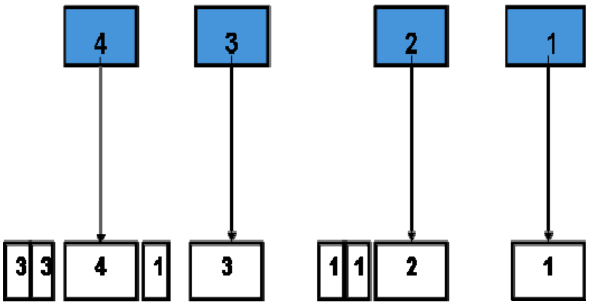


Fig. 3. Transmission of Replicas

Like Basic PR-TCP, Single-Side PR-TCP inherits congestion control mechanism from TCP NewReno except the retransmission of lost Regular packets. Pseudo packets will be retransmitted when a Regular packet is found lost. The state transition table is shown in Table 2.

Table 2. State Transition of Single-Side PR-TCP
(CWND: Congestion Window Size)

Event	Current State	Next State	Action
new ACK	(SS)	(SS)	CWND=CWND*2
Timeout	(SS)	(SS)	threshold=(1/2)*CWND, CWND=1
CWND>= threshold	(SS)	(CA)	CWND=CWND+1
new ACK	(CA)	(CA)	CWND=CWND+1
Timeout	(CA)	(SS)	CWND=1
triple duplicated ACKs	(CA)	(FF)	threshold = CWND*(1/2), CWND = threshold
duplicated ACK	(FF)	(FF)	No change
new ACK	(FF)	(CA)	CWND=CWND+3
Timeout	(FF)	(SS)	CWND=1

IV. PERFORMANCE EVALUATION

We use NS-2 network simulator [8] to evaluate Partial-Reliable TCP under various conditions such as link reliability, network size, and link bandwidth as well as delay bound. Two video samples are used on the EvalVid video quality evaluation system [4,7,9]. EvalVid is a framework and tool-set for the evaluation of the quality of video transmitted over a real or simulated communication network.

4.1 Evaluation Metrics

Three parameters are evaluated: Number of Packets in Decodable Frames (NPDF), Peak Signal to Noise Ratio (PSNR), and Wasted Transmission Overhead (WTO).

A. Number of Packets in Decodable Frames (NPDF)

A frame may be decomposed into several packets in transmission. The number of received packets must exceed a threshold for a frame to be decodable. Furthermore, the decodability of B-frames and P-frames depends on the decodability of other frames. Received packets that belong to non-decodable frames are useless. Therefore, NPDF is a useful evaluation parameter.

B. Peak Signal to Noise Ratio (PSNR)

PSNR is also a popular evaluation parameter for video quality. Its definition is shown in (1) and (2), where *MAX* is the largest value of a pixel, *I* and *K* are *m* by *n* frames, and MSE stands for Mean Square Error.

$$PSNR = 10 \cdot \log_{10}\left(\frac{MAX_I^2}{MSE}\right) = 20 \cdot \log_{10}\left(\frac{MAX_I}{\sqrt{MSE}}\right) \quad (1)$$

$$MSE = \frac{1}{mn}\sum_{i=0}^{m-1}\sum_{j=0}^{n-1}\left\| I(i,j) - K(i,j) \right\|^2 \quad (2)$$

Table 3 is the mapping between PSNR and Mean Opinion Score (MOS).

Table 3. PSNR vs. MOS

PSNR[db]	MOS
> 37	5 (Excellent)
31-37	4 (good)
25-31	3 (Fair)
20-25	2 (Poor)
	1 (Bad)

C. Wasted Transmission Overhead (WTO)

Overdue packets are useless for those applications that have stringent delay constraint. The number of packets that are received by the receiver but are overdue is another useful evaluation parameter.

WTO = (no. of Invalid packets
+ no. of retransmitted packets) x packet size (3)

4.2 Design of Experiments

Experiments are executed under both wired (Experiment A) and wired-wireless-hybrid (Experiment B) network environments. Two sample videos, Foreman and Container, are used. Traffic bursts are injected into the network to simulated different network conditions. Both versions of PR-TCP under various protection plans (protect I-frame only and protect I-/P-frames) are evaluated against TCP Reno, UDP, and TFRC [2].

- **PR(I)**: Basic PR-TCP with I-frames Certified, and Regular for others.
- **PR(I+P)**: Basic PR-TCP with I-frames and P-frames Certified, and Regular for B-frames.
- **PR-SS**: Single-Side PR-TCP with I-frames Registered, and Regular for others.

Common parameters used in both experiments are shown in Table 4.

Table 4. Parameters of Simulation A and B

Parameter	Range
No. of hops	2~10
Delay bound	0.6 sec~8 sec
Queue size	20 packets
Loss rate	0~0.5
Burst traffic load	500Kbps
Certified class	I-frame, I/P-frame

4.3 Experiment A: Wired Network

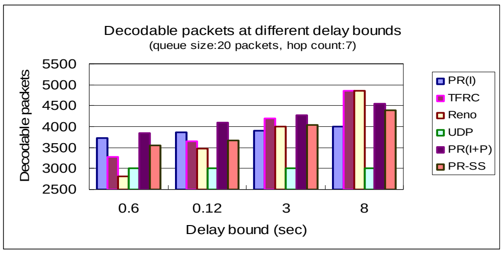
The topology used in Experiment A is shown in Fig. 4 and parameters are listed in Table 5. The simulation time is 15 seconds. Traffic bursts (500kbps) were injected into the network at the 4th and the 9th seconds. Link bandwidth is 5Mbps.

Fig. 4. Topology of Experiment A

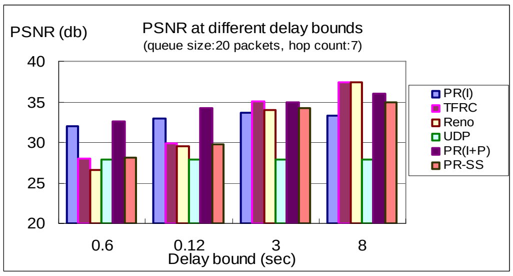
Table 5. Parameters of Simulation A

Parameter	Range
No. of hops	2~10
Delay bound	0.6 sec~8 sec
Queue size	20 packets
Certified class	I-frame, I&P-frame

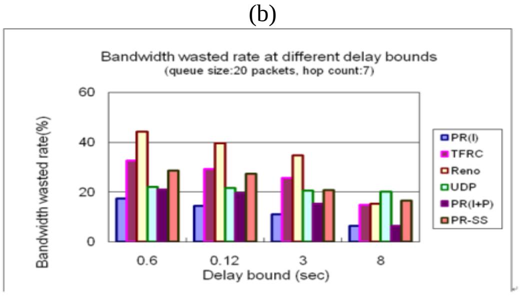
The two different video samples do not make much difference in the experiments. Only the experiments that use Foreman are shown in this paper. From Fig. 5 we can see that PR-TCP clearly outperforms others. UDP performs poorly in all delay bounds, either small or large. When the delay bound is large (>= 8 sec.), conventional TCPs are slightly better than PR-TCP, However, when the delay bound is small (0.6 sec.), Basic PR-TCP can outperform TCP Reno and TFRC in the NPDF by at least 18%. It outperforms TCP Reno, TFRC, and UDP in PSNR by at least 12%. The performance of Single-Side PR-TCP is lower than Basic PR-TCP in terms of PSNR by 10%, and it consumes 8% more bandwidth. Fig. 5(d) shows the tradeoff between bandwidth overhead and PSNR. When the delay bound is large, PR-TCP can waste a little bandwidth to gain a significant improvement in video quality. When the delay bound is small, PR-TCP not only can improve video quality, it can also save some bandwidth consumption because it wastes less bandwidth (WTO) than NewReno (and other TCPs).



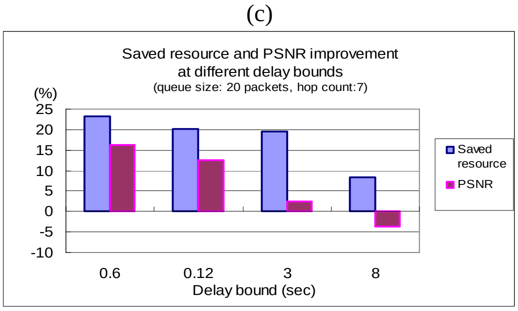
(a)



(b)



(c)



(d)

Fig. 5. Results of Experiment A: (a) Decodable Packets, (b) PSNR, (c) WTO, (d) PSNR Improvement vs. WTO Reduction.

Fig. 6 shows the frames abstract from the original video and the received videos. The original frame is frame #111 transmitted by the sender at the 6th second when a traffic burst was injected into the network.

(a) (b)

(c) (d)

(e) (f)

Fig. 6. Frames extracted from videos: (a) Original, (b) by TCP Reno, (c) by UDP, (d) by TFRC, (e) by Single-Side PR-TCP, (f) by Basic PR-TCP.

4.4 Experiment B: Wired and Wireless Hybrid Network

The wired and wireless hybrid topology used in Experiment B is shown in Fig. 7, in which, the last link is wireless with relative higher error rate. The parameters used in the experiment are listed in Table 6. Other simulation set-up is similar to Experiment A.

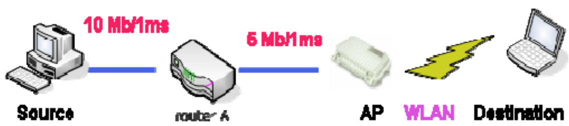
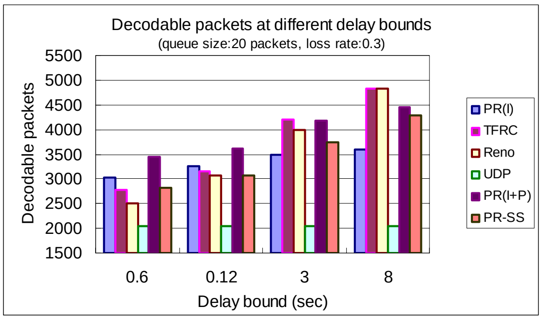


Fig. 7. Topology used in Experiment B

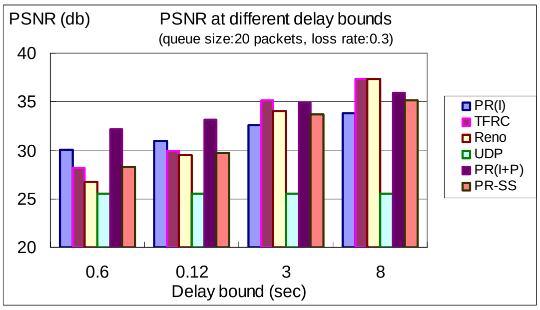
Table 6. Parameters of Simulation B

Parameter	Range
No. of hops	4
Delay bound	0.6 sec~8 sec
Queue size	20 packets
Loss rate	0~0.5
Certified class	I-frame, I&P-frame

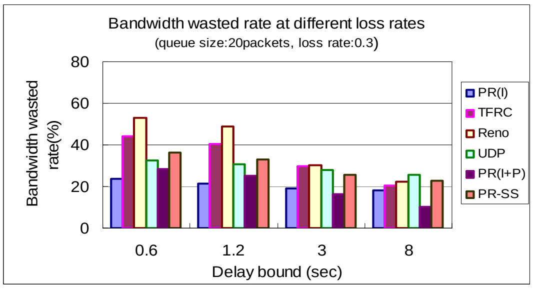
Similar to Experiment A, the two different video samples do not make much difference in the experiments. Likewise, only the experiments that use Foreman are shown. From Fig. 8 we can see that, similar to wired environment, PR-TCP clearly outperforms others. UDP still performs poorly in all delay bounds. When the delay bound is small (8 sec.), the quality (PSNR) of the video transmitted using Basic PR-TCP is downgraded by only 3% as compared to NewReno, while it consume 8% less bandwidth. The performance of Single-Side PR-TCP is about the same as Basic PR-TCP in terms of PSNR, but it consumes 5% more bandwidth.



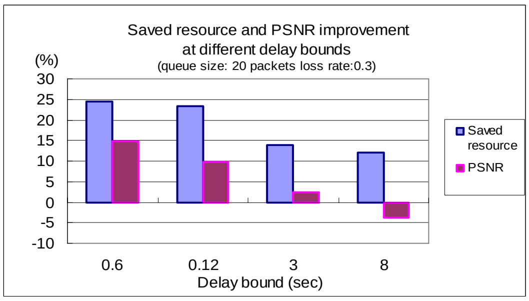
(a)



(b)



(c)



(d)

Fig. 8. Results of Experiment B: (a) Decodable Packets, (b) PSNR, (c) WTO, (d) PSNR Improvement vs. WTO Reduction.

V. CONCLUDING REMARKS

This paper proposes a new TCP protocol, Partial-Reliable TCP, which supports a selective packet protection plan to the applications that can tolerate packet loss. For those packets that are less important than others, an application can choose not to protect them without worrying significant quality degradation. In return, system and network resources can be saved and packets may be delivered faster. For those applications that have stringent delay constraints, faster packet delivery may result in higher packet survivability and better reply quality. Basic PR-TCP requires both ends of a connection to adopt PR-TCP while Single-Side PR-TCP only requires the sender side to adopt it. Our experiments over NS-2 Simulator demonstrate the superiority of PR-TCP.

REFERENCES

1. S. Floyd, T. Henderson, "The NewReno Modification to TCP's Fast Recovery Algorithm," *IETF RFC 2582*, 1999.
2. M. Handley, S. Floyd, J. Padhye, and J. Widmer, "TCP friendly rate control protocol specification (TFRC)," *RFC3448*, Jan. 2003.
3. V. Jacobson, "Congestion Avoidance and Control," *Proc. of ACM SIGCOMM*, Aug. 1988, pp. 314-329.
4. J. Klaue, B. Rathke, and A. Wolisz, "EvalVid - A Framework for Video Transmission and Quality Evaluation," *Proc. of the 13th Int'l Conf. on Modeling Techniques and Tools for Computer Performance Evaluation*, Sep. 2003, pp. 255-272.
5. J. Postel, "Transmission Control Protocol," *IETF RFC 793*, 1981.
6. J. Postel, "User Datagram Protocol," *IETF RFC 768*, 1981.
7. C.Y. Yu, C.H. Ke, C.K. Shieh, and N. Chilamkurti, "MyEvalvid-NT - A Simulation Tool-set for Video Transmission and Quality Evaluation," *TENCON 2006*, Nov. 2006, pp. 1-4.
8. "The Network Simulator - ns-2,"*http://www.isi.edu/nsnam/index.php/Main_Page/.*
9. "YUV Video Sequences," *http://trace.eas.asu.edu/yuv/index.html*.
10. ISO/IEC Int'l Standard 11172 (MPEG-1 -H.261); "Coding of moving pictures and associated audio for digital storage media up to about 1.5 Mbits/s," *1993.*
11. *ISO/IEC Int'l Standard 13818 (MPEG-2 -H.262); "Generic coding of moving pictures and associated audio information," 1994.*